CONFERENCING SYSTEM

CROSS-REFERENCE TO RELATED APPLICATIONS

[0001] This application is a continuation of copending U.S. patent application Ser. No. 12/838,947 filed on Jul. 19, 2010, which is a continuation of U.S. patent application Ser. No. 10/645,848 filed on Aug. 22, 2003, both of which are incorporated herein by reference in their entirety.

FIELD OF THE INVENTION

[0002] The present invention relates to a method and apparatus for setting up conference calls in communication systems, and in particular but not exclusively to wireless communication systems.

BACKGROUND OF THE INVENTION

[0003] The concept of conference calls in public switched telephone networks (PSTN) is well known. PSTN conferences are typically set up by a first participant calling a specific customer support number and being supplied with a conference bridge number and a PIN code. The first participant can then provide this information to any other potential participants. The participants wishing to join the call would each dial the conference bridge number, and supply the PIN code on demand, and would subsequently be admitted to the conference call.

[0004] As an alternative, the Internet could conceivably be used to arrange conference calls. A specific web site could be accessed by a first participant, and a bridge number and PIN code could be obtained. The first participant would then be able to provide the details to other participants.

[0005] Both of these procedures allow for a mobile terminal to be involved in the conference call. However, both procedures have two main disadvantages. Firstly, a conference call must be planned in advance. The various participants must contact each other so that they each know when the call is due to take place and can dial the conference bridge number at that time. Secondly, the participants themselves need to organize for the bridge number and the PIN code to be distributed to all participants.

[0006] Various models have been proposed for providing conferencing services in third generation Internet Protocol Multimedia Subsystem (IMS) wireless communication systems, including an IETF draft entitled, "Models for Multi Party Conferencing in SIP", by J. Rosenberg and H. Schulzrinne. Each of the models in this draft uses Session Initiation Protocol (SIP) messaging. The SIP protocol is discussed in Internet Standards RFC 3261 and RFC 2543. Some of the models are described briefly hereinafter.

[0007] The first model, known as "end system mixing", requires that one terminal involved in a conference call performs the mixing (merging) of signals and media streams sent to and from other terminals in the call. FIG. **1**A is a depiction of a three-way call using this model. In this example, users A and B are involved in a two-way call. At some point during the call, user A decides to bring user C into the call. To do this, user A calls user C using a completely separate SIP call. There is no call set up between B and C. Instead, A receives media streams from both B and C and mixes them. Terminal A sends a stream containing the streams of A and B to terminal C, and a stream containing A's and C's streams to terminal B. In this model, terminals B and C are unaware from a SIP perspective that the call involves more than two parties.

[0008] In the case of a call involving more than three terminals, more than one terminal may perform mixing and signalling to sustain the call. For instance, as an extension of the above-described example, user C may decide to invite a fourth user D into the conference call. User C would then call user D and terminal C would perform the mixing of the streams it receives from terminal A with its own stream, and send the combined stream to D, and mix its own stream with that of D and send this to A. This set-up is shown in FIG. **1**B.

[0009] Serious disadvantages of this model are that when a mixing terminal leaves the call, the conference must end, and that there is no way for a mixing terminal to determine whether a signalling message sent to it was intended for that terminal alone or for all terminals in the conference.

[0010] A further model, using dial-in conference servers, closely mirrors the PSTN system described above. One participant defines a URI (uniform resource identifier) to identify a conference call, and sends it to other participants. The participants then each call the server, using the conference URI, which maintains point-to-point SIP relationships with each participant that calls in. The server receives media from each participant, mixes them, and sends out the appropriate mixed stream to each participant separately. This model is depicted in FIG. **2**, which shows four users A-D taking part in a conference call.

[0011] Dial-in conference servers are versatile in that they can be used for pre-arranged conferences or for ad hoc conferences. However, this model suffers from the fact that it is possible for the same URI to be used for more than one conference. This would cause conference sessions to be mixed.

[0012] It is an object of the present invention to provide a solution to one or more of the previously-stated problems.

SUMMARY OF THE INVENTION

[0013] According to a first aspect of the present invention, a method is provided for a first user equipment to administer conferencing resources in a communications system comprising at least one other user equipment and a server, the method comprising: transmitting from the first user equipment to the server a first message comprising a request for a resource configured to sustain a conference call; receiving by the first user equipment from the server a second message comprising a network address identifying the resource configured to sustain the conference call which has been allocated by the server; in response to receiving the second message, transmitting a first request from the first user equipment directly to the resource at the network address; in response to receiving an acknowledgment of the first request directly from the resource, transmitting from the first user equipment to at least one other user equipment a third message comprising the network address; and in response to receiving a notification that the resource sends out directly to the at least one other user equipment an acknowledgment of a second request directly sent from the at least one terminal, the first user equipment initiating a connection from the first user equipment to the at least one other user equipment via the resource to establish a conference call between the first user equipment and the at least one other user equipment; wherein the third message comprising the